Amendments to the Claims

Please amend Claims 1, 9, 25, 31, and 55. The Claim Listing below will replace all prior versions of the claims in the application:

Claim Listing

1. (Currently Amended) In a communications system for transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics including a first characteristic, said first parameter being related to said first characteristic, said compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said parameters related to said first characteristic, apparatus for adjusting the first characteristic comprising:

~~a transmitter transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter,~~

~~wherein said parameters represent an audio signal comprising a plurality of audio characteristics including a first characteristic,~~

~~wherein said first parameter is related to said first characteristic,~~

~~wherein said compression code is decodable by a plurality of decoding steps including a first decoding step for decoding said parameters related to said first characteristic; and~~

a processor responsive to said digital signals and said compression code to at least partially decode said digital signals to read at least said first parameter and to generate at least a first parameter value derived from said first parameter, responsive to said digital signals and said first parameter value to generate an adjusted first parameter value representing an adjustment of said first characteristic, and responsive to said adjusted first parameter value to derive an adjusted first parameter and to replace said first parameter with said adjusted first parameter[[,]].

~~wherein said digital signals are adapted to originate at a first handset and be received at a second handset and said processor is located away from said first handset and said second handset.~~

2. (Original) Apparatus, as claimed in claim 1, wherein said first characteristic comprises a level of said audio signal.

3. (Original) Apparatus, as claimed in claim 1, wherein said plurality of decoding steps further comprise at least one decoding step avoiding substantial altering of the first characteristic and wherein said processor avoids performing said at least one decoding step.

4. (Original) Apparatus, as claimed in claim 3, wherein said at least one decoding step comprises post-filtering.

5. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises a linear predictive code.

6. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises regular pulse excitation – long term prediction code.

7. (Original) Apparatus, as claimed in claim 6, wherein said digital signals are transmitted in frames comprising subframes and wherein said first parameter comprises a maximum absolute value of the elements in a codebook vector for one of said subframes.

8. (Original) Apparatus, as claimed in claim 1, wherein said compression code comprises algebraic code-excited linear prediction code.

9. (Currently Amended) Apparatus, as claimed in claim 8, wherein said digital signals are transmitted in frames comprising subframes, wherein said first parameter comprises a gain correction factor for one <u>of said</u> subframes.

10. (Original) Apparatus, as claimed in claim 1, wherein said digital signals comprise a near end digital signal using a near end compression code comprising a predetermined plurality of near end parameters including a first near end parameter, said near end parameters representing a near end audio signal comprising a plurality of near end audio characteristics including a near end first characteristic, said near end first parameter being related to said near end first characteristic, said near end compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said near end parameters related to said near end first characteristic, said digital signals further comprising a far end digital signal using a far end compression code comprising a predetermined plurality of far end parameters, said far end parameters representing a far end audio signal comprising a plurality of far end audio characteristics including a far end first characteristic, said far end compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said far end parameters related to said far end first characteristic,

wherein said processor receives said near end digital signal and said far end digital signal,

wherein said processor is responsive to said near end digital signal to read at least said near end first parameter and to generate a near end first parameter value derived from said near end first parameter,

wherein said processor is responsive to said near end digital signal to perform at least said first decoding step to generate near end decoded signals related to said near end first characteristic of said near end audio signal,

wherein said processor is responsive to said far end digital signal to perform at least said first decoding step to generate far end decoded signals related to said far end first characteristic of said far end audio signal,

wherein said processor is responsive to said near end decoded signals, said far end decoded signals and said near end first parameter value to generate an adjusted near end first parameter value representing an adjustment of said near end first characteristic,

wherein said processor derives an adjusted near end first parameter from said adjusted near end first parameter value, and

wherein said processor replaces said near end first parameter with said adjusted near end first parameter.

11. (Original) Apparatus, as claimed in claim 1, wherein said processor tests said adjusted first parameter value for an overflow and underflow condition before deriving said adjusted first parameter.

12. (Original) Apparatus, as claimed in claim 11, wherein said first parameter is a quantized first parameter and wherein said processor derives said adjusted first parameter by quantizing said adjusted first parameter value.

13. (Original) Apparatus, as claimed in claim 12, wherein said processor uses differential scalar quantization during said quantizing.

14. (Previously presented) Apparatus, as claimed in claim 13, wherein said processor uses differential scalar quantization with a quantizer outside a feedback loop during said quantizing.

15. (Original) Apparatus, as claimed in claim 1, wherein said first parameter comprises a series of first parameters received over time, wherein said processor is responsive to said digital signals to read said series of first parameters and to generate a series of first parameter values over time, and wherein said processor is responsive to said decoded signals and to at least a plurality of said series of first parameter values to generate said adjusted first parameter value.

16. (Original) Apparatus, as claimed in claim 15, wherein said first parameter is a quantized first parameter and wherein said processor derives said adjusted first parameter by quantizing said adjusted first parameter value.

17. (Original) Apparatus, as claimed in claim 16, wherein said processor uses differential scalar quantization during said quantizing.

18. (Original) Apparatus, as claimed in claim 1, wherein said first parameter is a quantized first parameter and wherein said processor derives said adjusted first parameter by quantizing said adjusted first parameter value.

19. (Original) Apparatus, as claimed in claim 18, wherein said processor uses differential scalar quantization during said quantizing.

20. (Original) Apparatus, as claimed in claim 18, wherein said processor performs said quantizing using instantaneous scalar quantization techniques.

21. (Original) Apparatus, as claimed in claim 1, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said processor is responsive to said digital signals to read at least said first parameter from each of said plurality of subframes, and wherein said processor replaces said first parameter with said adjusted first parameter in each of said plurality of subframes.

22. (Original) Apparatus, as claimed in claim 21, wherein said processor replaces said first parameter with said adjusted first parameter for a first subframe before processing a subframe following the first subframe to achieve lower delay.

23. (Original) Apparatus, as claimed in claim 1, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said processor performs at least said first decoding step during a first of said subframes to generate said decoded signals, reads said first parameter from a second of said subframes occurring subsequent to said first subframe to generate said first parameter value, generates said adjusted first parameter value in response to said decoded signals

and said first parameter value, and replaces said first parameter of said second subframe with said adjusted first parameter.

24. (Original) Apparatus, as claimed in claim 1, wherein said processor performs at least said first decoding step to generate decoded signals related to said first characteristic of said audio signal and wherein said processor is responsive to said decoded signals and said first parameter value to generate said adjusted first parameter value.

25. (Currently Amended) In a communications system for transmitting digital signals comprising code samples, said code samples comprising first bits using a compression code and second bits using a linear code, said code samples representing an audio signal, said audio signal having a plurality of audio characteristics including a first characteristic, apparatus for adjusting the first characteristic comprising:

~~a first transmitter transmitting digital signals comprising code samples,~~

~~wherein said code samples comprise first bits using a compression code and second bits using a linear code,~~

~~wherein said code samples represent a first audio signal,~~

~~wherein said first audio signal has a plurality of audio characteristics including a first characteristic;~~

~~a second transmitter transmitting digital signals comprising code samples,~~

~~wherein said code samples comprise third bits using a compression code and fourth bits using a linear code,~~

~~wherein said code samples represent a second audio signal,~~

~~wherein said second audio signal has a plurality of audio characteristics including a second characteristic~~

~~wherein said second audio signal is from a source other than the source of said first audio signal; and~~

a processor responsive to said second bits ~~and at least one of said third bits and said fourth bits~~ to adjust said first bits and said second bits[[,]] ~~whereby~~ and, in turn, adjust said first characteristic ~~is adjusted~~[[,]] without decoding said compression code; and

~~wherein said processor adjusts said first characteristic without decoding said compression code.~~ a transmitter to transmit digital signals with adjusted first bits and second bits to a device to produce a corresponding audible signal with the first characteristic in an adjusted state.

26. (Original) Apparatus, as claimed in claim 25, wherein said linear code comprises pulse code modulation (PCM) code.

27. (Original) Apparatus, as claimed in claim 25, wherein said first characteristic comprises audio level.

28. (Original) Apparatus, as claimed in claim 25, wherein said compression code samples conform to the tandem-free operation of the global system for mobile communications standard.

29. (Original) Apparatus, as claimed in claim 25, wherein said first bits comprise the two least significant bits of said samples and wherein said second bits comprise the 6 most significant bits of said samples.

30. (Original) Apparatus, as claimed in claim 29, wherein said 6 most significant bits comprise PCM code.

31. (Currently Amended) In a communications system for transmitting digital signals using a compression code comprising a predetermined plurality of parameters including a first parameter, said parameters representing an audio signal comprising a plurality of audio characteristics including a first characteristic, said first parameter being related to said first characteristic, said compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said parameters related to said first characteristic, a method of adjusting the first characteristic comprising:

reading at least said first parameter in response to said digital signals;

generating at least a first parameter value derived from said first parameter;

performing at least said first decoding step to generate decoded signals related to said first characteristic of said audio signal in response to said digital signals;

generating an adjusted first parameter value representing an adjustment of said first characteristic in response to said digital signals and said first parameter value;

deriving an adjusted first parameter in response to said adjusted first parameter value; and

replacing said first parameter with said adjusted first parameter[[,]].

~~wherein said digital signals are adapted to originate at a first handset and be received at a second handset and said replacing takes place away from said first handset and said second handset.~~

32. (Original) A method, as claimed in claim 31, wherein said first characteristic comprises a level of said audio signal.

33. (Original) A method, as claimed in claim 31, wherein said plurality of decoding steps further comprise at least one decoding step avoiding substantial altering of the first characteristic and wherein said method avoids performing said at least one decoding step.

34. (Original) A method, as claimed in claim 33, wherein said at least one decoding step comprises post-filtering.

35. (Original) A method, as claimed in claim 31, wherein said compression code comprises a linear predictive code.

36. (Original) A method, as claimed in claim 31, wherein said compression code comprises regular pulse excitation – long term prediction code.

37. (Original) A method, as claimed in claim 36, wherein said digital signals are transmitted in frames comprising subframes and wherein said first parameter comprises a maximum absolute value of the elements in a codebook vector for one of said subframes.

38. (Original) A method, as claimed in claim 31, wherein said compression code comprises code-excited linear prediction code.

39. (Original) A method, as claimed in claim 38, wherein said digital signals are transmitted in frames comprising subframes, wherein said first parameter comprises a gain correction factor.

40. (Original) A method, as claimed in claim 31, wherein said digital signals comprise a near end digital signal using a near end compression code comprising a predetermined plurality of near end parameters including a first near end parameter, said near end parameters representing a near end audio signal comprising a plurality of near end audio characteristics including a near end first characteristic, said near end first parameter being related to said near end first characteristic, said near end compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said near end parameters related to said near end first characteristic, said digital signals further comprising a far end digital signal using a far end compression code comprising a predetermined plurality of far end parameters, said far end parameters representing a far end audio signal comprising a plurality of far end audio characteristics including a far end first characteristic, said far end compression code being decodable by a plurality of decoding steps including a first decoding step for decoding said far end parameters related to said far end first characteristic,

wherein said receiving said digital signals comprises receiving said near end digital signal and said far end digital signal,

wherein said reading comprises reading at least said near end first parameter;

wherein said generating a first parameter comprises generating a near end first parameter value derived from said near end first parameter,

wherein said performing at least said first decoding steps comprises generating near end decoded signals related to said near end first characteristic of said near end audio signal in

response to said near end digital signal and generating far end decoded signals related to said far end first characteristic of said far end audio signal in response to said far end digital signal,

wherein said generating an adjusted first parameter value comprises generating an adjusted near end first parameter value representing an adjustment of said near end first characteristic in response to said near end decoded signals, said far end decoded signals and said near end first parameter value,

wherein said deriving an adjusted first parameter comprises deriving an adjusted near end first parameter from said adjusted near end first parameter value, and

wherein said replacing comprises replacing said first parameter with said adjusted first parameter.

41. (Original) A method, as claimed in claim 31, and further comprising testing said adjusted first parameter value for an overflow and underflow condition before deriving said adjusted first parameter.

42. (Original) A method, as claimed in claim 41, wherein said first parameter is a quantized first parameter and wherein said deriving an adjusted first parameter comprises quantizing said adjusted first parameter value.

43. (Original) A method, as claimed in claim 42, and further comprising using differential scalar quantization during said quantizing.

44. (Previously presented) A method, as claimed in claim 43, wherein said using differential scalar quantization comprises using a quantizer outside a feedback loop during said quantizing.

45. (Original) A method, as claimed in claim 31, wherein said first parameter comprises a series of first parameters received over time, wherein said reading at least said first parameter comprises reading said series of first parameters, wherein said generating a first parameter value comprises generating a series of first parameter values over time, and wherein said generating an adjusted first parameter value comprises generating said adjusted first parameter value in response to said decoded signals and to at least a plurality of said series of first parameter values.

46. (Original) A method, as claimed in claim 45, wherein said first parameter is a quantized first parameter and wherein said deriving an adjusted first parameter comprises quantizing said adjusted first parameter value.

47. (Original) A method, as claimed in claim 46, and further comprising using differential scalar quantization during said quantizing.

48. (Original) A method, as claimed in claim 31, wherein said first parameter is a quantized first parameter and wherein said deriving an adjusted first parameter comprises quantizing said adjusted first parameter value.

49. (Original) A method, as claimed in claim 48, and further comprising using differential scalar quantization during said quantizing.

50. (Original) A method, as claimed in claim 48, wherein said quantizing comprises using instantaneous scalar quantization techniques.

51. (Original) A method, as claimed in claim 31, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said reading at least said first parameter comprises reading at least said first parameter from each of said plurality of subframes, and wherein said replacing comprises replacing said first parameter with said adjusted first parameter in each of said plurality of subframes.

52. (Original) A method, as claimed in claim 51, wherein said replacing comprises replacing said first parameter with said adjusted first parameter for a first subframe before processing a subframe following the first subframe to achieve lower delay.

53. (Original) A method, as claimed in claim 31, wherein said compression code is arranged in frames of said digital signals and wherein said frames comprise a plurality of subframes each comprising said first parameter, wherein said performing at least said first decoding step comprises performing at least said first decoding step during a first of said subframes to generate said decoded signals, wherein said reading at least said first parameter comprises reading at least said first parameter from a second of said subframes occurring subsequent to said first subframe, wherein said generating a first parameter value comprises generating a first parameter value from said first parameter from said second of said subframes, wherein said generating an adjusted first parameter value comprises generating said adjusted first parameter value in response to said

decoded signals and said first parameter value, and wherein said replacing comprises replacing said first parameter of said second subframe with said adjusted first parameter.

54. (Original) A method, as claimed in claim 31, wherein said generating an adjusted first parameter comprises performing at least said first decoding step to generate decoded signals related to said first characteristic of said audio signal in response to said compression code and wherein said generating an adjusted first parameter is responsive to said decoded signals and said first parameter value.

55. (Currently Amended) In a communications system for transmitting digital signals comprising code samples, said code samples comprising first bits using a compression code and second bits using a linear code, said code samples representing an audio signal, said audio signal having a plurality of audio characteristics including a first characteristic, a method of adjusting ~~a signal~~ the first characteristic comprising:

~~receiving digital signals comprising code samples, wherein said code samples comprise first bits using a compression code and second bits using a linear code,~~

~~wherein said code samples represent a first audio signal,~~

~~wherein said first audio signal has a plurality of audio characteristics including a first characteristic;~~

~~receiving digital signals comprising code samples, wherein said code samples comprise third bits using a compression code and fourth bits using a linear code,~~

~~wherein said code samples represent a second audio signal,~~

~~wherein said second audio signal has a plurality of audio characteristics including a second characteristic,~~

~~wherein said second audio signal is from a source other than the source of said first audio signal; and~~

adjusting said first bits and said second bits in response to said second bits ~~and at least one of said third bits and said fourth bits, whereby~~ to adjust said first characteristic ~~is adjusted in order to control the noise characteristic of said digital signals,~~ without decoding said compression code; and

~~wherein said adjusting takes place without decoding said compression code,~~

transmitting digital signals with adjusted first bits and second bits to a device to produce a corresponding audible signal with the first characteristic in an adjusted state.

56. (Original) A method, as claimed in claim 55, wherein said linear code comprises pulse code modulation (PCM) code.

57. (Original) A method, as claimed in claim 55, wherein said first characteristic comprises audio level.

58. (Original) A method, as claimed in claim 55, wherein said compression code samples conform to the tandem-free operation of the global system for mobile communications standard.

59. (Original) A method, as claimed in claim 55, wherein said first bits comprise the two least significant bits of said samples and wherein said second bits comprise the 6 most significant bits of said samples.

60. (Original) A method, as claimed in claim 59, wherein said 6 most significant bits comprise PCM code.